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EXAMINER	
CRUTCHFIELD, CHRISTOPHER M	

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NOTIFICATION DATE	DELIVERY MODE
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Notice of the Office communication was sent electronically on above-indicated "Notification Date" to the following e-mail address(es):

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Office Action Summary	**Application No.** 10/588,501	**Applicant(s)** KATO, MOTOKI
	Examiner Christopher Crutchfield	**Art Unit** 2466

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *04 August 2006*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-23* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-23* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *04 August 2006* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ ~~Information Disclosure Statement(s) (PTO/SB/08)~~ Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 101

1. 35 U.S.C. 101 reads as follows:

> Whoever invents or discovers any new and useful process, machine, manufacture, or composition of matter, or any new and useful improvement thereof, may obtain a patent therefor, subject to the conditions and requirements of this title.

2. **Claims 3, 7, 8, 9, 15 and 20-23** rejected under 35 U.S.C. 101 because they are drawn to non-statutory subject matter.

Regarding claims 3, 7, 15 and 20 the claims are rejected under 35 U.S.C. 101 because they are directed to non statutory subject matter. That is, claims 3, 7, 15 and 20 are directed to software per se, which is not a process, machine, manufacture or composition of matter.

In order to qualify as statutory subject matter, claimed subject matter must fall within one of the four statutory categories of 35 USC 101. In re Nuijten, 500 F3d 1346, 1354 84 USPQ2d 1495, 1500 (2007) ("Claimed subject matter must be within at least one of four categories enumerated in 35 U.S.C. §101 in order to be patentable, but once that requirement is satisfied, court need not be overly concerned about which of those categories claimed subject matter falls into; however, four categories in Section 101 together describe exclusive reach of patentable subject matter, and if claim covers material not found in any of those four categories, then claim falls outside plainly expressed scope of Section 101 even if subject matter is otherwise new and useful."). Software per se it is not a process, manufacture or composition of matter. Furthermore, since it lacks any physical manifestation it cannot comprise a machine. See *Id at*

1354. See also Ex Parte *Cherian*, Appeal No. 2008-004157, BPAI, (Non-Precedential) (2009); Ex Parte Magid, Appeal No. 2008-3824, BPAI, (Non-Precedential) (2009).

Since the "program" of claims 3, 7, 15 and 20 only performs processing on an input stream (See Claims 3, 7, 15 and 20, Lines 1-4, 1-7, 1-4 and 2-5, Respectively), under its broadest reasonable interpretation the program of claims 3, 7, 15 and 20 comprises software per se. Therefore, under the broadest reasonable interpretation, claims 3, 7, 15 and 20 are directed non-statutory subject matter.

Regarding claims 8, 9 and 21-23 the claims are rejected under 35 U.S.C. 101 because they are directed to non statutory subject matter. That is, claims 8, 9, and 23 are directed to a data structure (See claims 8, 9, and 21-23, line 1) which is not among the four categories of statutory subject matter.

In order to qualify as statutory subject matter, claimed subject matter must fall within one of the four statutory categories of 35 USC 101. In re Nuijten, 500 F3d 1346, 1354 84 USPQ2d 1495, 1500 (2007) ("Claimed subject matter must be within at least one of four categories enumerated in 35 U.S.C. §101 in order to be patentable, but once that requirement is satisfied, court need not be overly concerned about which of those categories claimed subject matter falls into; however, four categories in Section 101 together describe exclusive reach of patentable subject matter, and if claim covers material not found in any of those four categories, then claim falls outside plainly expressed scope of Section 101 even if subject matter is otherwise new and useful."). A data structure it is not a process, manufacture, composition of matter or machine. See *Id at 1354*. Therefore, claims 8, 9 and 21-23 are rejected under 25 USC 101 as being directed to non-statutory subject matter.

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. The factual inquiries set forth in *Graham* v. *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

3. **Claims 1-3, and 8** are rejected under 35 U.S.C. 103(a) as being unpatentable over The ISO/IEC 13818-1 Standard ("The Standard") (Author Unknown, Generic Coding of Moving Pictures and Associated Audio: Systems, International Organization for Standardization, Workgroup 11 - Coding of Moving Pictures and Associate Audio, Pages 1-130, 13 November 1994) in view of *Bruls*, et al., (US Pre Grant Publication No. 2006/0098937 A1).

Regarding claims 1-3, *The Standard* discloses an information processing apparatus, method and computer to execute processing comprising:

a. An Encoding means and step for encoding an input stream so as to include, among a first to nth elementary streams associated with the input stream, a first elementary stream (Pages xi-xix, 3-6, 10-13, 21-22,43-50). (The system of The Standard discloses a system for the encoding and transport of MPEG data [Pages xi-xix]. The system operates by receiving an incoming video stream [i.e. input stream] [See Fig. 0-1, "Video data" and "Audio Data", Page xi], encoding the stream into one or more packetized elementary streams ["PES"] associated with the input stream [Pages xi-xii] [See also Page 35, Table 2-19, "ISO/IEC 12818-3 ... audio steam number" - Showing the elementary streams may be part of a layered audio architecture], packetizing the one or more PES into a transport stream ["TS"] [Fig. 0-1, "Packetizer" and "Mux", Page xii] [See also Pages xvi-xix, particularly sections 0.4, 0.7 and 0.8], transmitting the TS across a network to a decoder [Page xii], receiving the TS at the decoder, separating out the input stream by de-multiplexing the PES's associated with the input stream, decoding the input stream and providing the decoded video/audio for output [Pages xii-xiii, Particularly Fig. 0-2] [See also Pages xvi-xix, Particularly Sections 0.4, 0.7 and 0.8].)

b. Table generating means and step for generating a table in which information is written for associating IDs that respectively identify the first to nth elementary streams, which are encoded by the encoding means, with the first to nth elementary streams (Pages 10-17, 22 and 43-50). (The system of *The Standard* further discloses that each packetized elementary stream is assigned a packet identifier ["PID"] [i.e. IDs] that is used to uniquely identify that stream in the transport stream [TS] [See Particularly Section 2.4.1, Page 10 and "PID", Page 22]. Within each TS, a program association table and program map table are periodically transmitted in a special PES packets [Pages 43-50,

Particularly Section 4.3.3 on Pages 43-44]. The program association table associates a particular program with a program map table ID, and the program map table associates the PIDs of the elementary streams that make up a program with the program map table ID [Pages 43-50]. Therefore the program map table associates an ID with each of the first to nth elementary streams.)

c. Adding means and step for adding the corresponding IDs to the first to nth elementary streams encoded by the encoding means (Fig. 0-1, “Packetizer” and "Mux", Page xii, and “PID”, Page 22). (The Standard discloses that each packet of the PES bears the PID associated with that elementary stream).

d. Packetizing means and step for packetizing the first to nth elementary streams, to which the IDs are added by the adding means, and the table into TS packets (Fig. 0-1, “Packetizer” and "Mux", Page xii, and “PID”, Page 22). (The Standard discloses that all of the individual streams, including the streams bearing the program association table and program map table are packetized, including adding the elementary stream ID to the packet, and multiplexed into a single TS [Fig. 0-1, “Packetizer” and "Mux", Page xii, Pages 22 and 43-46].)

The Standard fails to disclose that the first to nth elementary streams may comprise a base stream and a first to n-th extension streams such that the information processing apparatus, method and program comprises an encoding means and step for encoding an input stream so as to include, among a base stream and first to n-th extension streams having extensibility for the base stream, at least the base stream and the first extension stream, table

generating means and step for generating a table in which information is written for associating IDs that respectively identify the base stream and the first to n-th extension streams, which are encoded by the encoding means and step, with the base stream and the first to n-th extension streams, adding means and step for adding the corresponding IDs to the base stream and the first to n-th extension streams encoded by the encoding means and packetizing means and step for packetizing the base stream and the first to n-th extension streams, to which the IDs are added by the adding means and step, and the table into TS packets. In the same field of endeavor, *Bruls* discloses the first to nth elementary streams may comprise a base stream and a first to n-th extension streams such that the information processing apparatus, method and program comprises an encoding means and step for encoding an input stream so as to include, among a base stream and first to n-th extension streams having extensibility for the base stream, at least the base stream and the first extension stream, table generating means and step for generating a table in which information is written for associating IDs that respectively identify the base stream and the first to n-th extension streams, which are encoded by the encoding means and step, with the base stream and the first to n-th extension streams, adding means and step for adding the corresponding IDs to the base stream and the first to n-th extension streams encoded by the encoding means and packetizing means and step for packetizing the base stream and the first to n-th extension streams, to which the IDs are added by the adding means and step, and the table into TS packets (Paragraphs 0031, 0039, and 0042). (The system of *Bruls* discloses the use of a base and one or more extensible enhancement layers for encoding and transporting programs using MPEG encoding techniques [Paragraphs 0031 and 0039, See also Paragraph 0003]. *Bruls* further discloses that the system uses a separate PID to identify the base and enhancement layers [Paragraph 0042].)

Therefore, since *Bruls* discloses the use of a base layer and multiple enhancement layers and the use of a separate PID for each of the layers, and *The Standard* discloses the use of a separate elementary stream for each PID for different encoding layers and the use of a program map table to identify the association of each PID/elementary stream with a particular program, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the PID identified layers of *Bruls* with the system of *The Standard* by associating each of the base and enhancement layers with a PID, as taught by *Bruls*, assigning each of the PIDs to a particular elementary stream, as taught by *The Standard*, and transmitting a table identifying the PIDs in a transport stream packet, as taught by *The Standard*. The motive to combine is to allow the system of *The Standard* to support layered video, which allows increased flexibility and reduced bandwidth requirements for video distribution (See Generally, *Bruls*, Paragraphs 0006-0014).

Finally, assuming arguendo, that *Bruls* fails to disclose the use of more the one enhancement layers (i.e. Because *Bruls* is not in standard US format with a clearly identified background of the invention, it is unclear if Paragraph 0003 is a part of the background of the invention, therefore constituting a separate disclosure.) In the same field of endeavor, The Background of *Bruls* discloses the use of one or more enhancement layers (See Paragraph 0003).

Therefore, since the background of *Bruls* discloses the use of more then one extension layer (i.e. a first to n-the extension stream), it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the multiple extension layers of The Background of *Bruls* into the teachings of *The Specification* as modified by *Bruls* by transmitting more then one extension stream as a separate elementary stream with its own stream ID. The motive to combine is to allow enhanced flexibility by providing multiple extension streams so

that endpoints can decode one or more of the extension streams to obtain incremental quality increases.

Regarding claims 8, *The Standard* discloses a data structure of an entire stream to be played back by a computer, the entire stream including a first to nth elementary stream, wherein the entire stream includes:

a. TS packets forming the first stream, TS packets forming each of the second to n-th elementary streams (Pages xi-xix, 3-6, 10-13, 21-22,43-50). (The system of The Standard discloses a system for the encoding and transport of MPEG data [Pages xi-xix]. The system operates by receiving an incoming video stream [i.e. input stream] [See Fig. 0-1, "Video data" and "Audio Data", Page xi], encoding the stream into one or more packetized elementary streams ["PES"] associated with the input stream [Pages xi-xii] [See also Page 35, Table 2-19, "ISO/IEC 12818-3 ... audio steam number" - Showing the elementary streams may be part of a layered audio architecture], packetizing the one or more PES into a transport stream ["TS"] [Fig. 0-1, "Packetizer" and "Mux", Page xii] [See also Pages xvi-xix, particularly sections 0.4, 0.7 and 0.8], transmitting the TS across a network to a decoder [Page xii], receiving the TS at the decoder, separating out the input stream by de-multiplexing the PES's associated with the input stream, decoding the input stream and providing the decoded video/audio for output [Pages xii-xiii, Particularly Fig. 0-2] [See also Pages xvi-xix, Particularly Sections 0.4, 0.7 and 0.8].)

b. A TS packet storing a table in which information is written for associating the TS packets forming the first to n-th elementary streams with IDs that respectively identify the TS packets (Pages 10-17, 22 and 43-50). (The system of *The Standard* further discloses

that each packetized elementary stream is assigned a packet identifier ["PID"] [i.e. IDs] that is used to uniquely identify that stream in the transport stream [TS] [See Particularly Section 2.4.1, Page 10 and "PID", Page 22]. Within each TS, a program association table and program map table are periodically transmitted in a special PES packets [Pages 43-50, Particularly Section 4.3.3 on Pages 43-44]. The program association table associates a particular program with a program map table ID, and the program map table associates the PIDs of the elementary streams that make up a program with the program map table ID [Pages 43-50]. Therefore the program map table associates an ID with each of the first to nth elementary streams.)

c. A header of each of the TS packets forming each of the first to n-th elementary streams includes the ID identifying the TS packet (Fig. 0-1, “Packetizer” and "Mux", Page xii, and “PID”, Page 22). (The Standard discloses that all of the individual streams, including the streams bearing the program association table and program map table are packetized, including adding the elementary stream ID to the packet, and multiplexed into a single TS [Fig. 0-1, “Packetizer” and "Mux", Page xii, Pages 22 and 43-46].)

4. **Claim 9** is rejected under 35 U.S.C. 103(a) as being unpatentable over The ISO/IEC 13818-1 Standard (“The Standard”) (Author Unknown, Generic Coding of Moving Pictures and Associated Audio: Systems, International Organization for Standardization, Workgroup 11 - Coding of Moving Pictures and Associate Audio, Pages 1-130, 13 November 1994) and *Bruls*, et al., (US Pre Grant Publication No. 2006/0098937 A1) as applied to claims XXX and further in

view of *Kim*, et al. (S. Kim, S. Park, Y. Kim, Fine Grain Scalability in MPEG-4 Audio, Audio Engineering Society, 111th Convention of The AES, 24 Sept 2001, Pages 1-5).

Regarding claim 9, The Standard fails to disclose a data structure wherein the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams, which are included in the entire stream, are arranged in sequence of the TS packets to be played back at the same time and in the order of the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams. In the same field of endeavor, *Kim* discloses a data structure wherein the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams, which are included in the entire stream, are arranged in sequence of the TS packets to be played back at the same time and in the order of the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams (Page 3, Fig. 3). (The system of Kim discloses a system where the base and enhancement layers of a transmitted bit stream for a timeslot/frame are multiplexed in layer order with the base stream first, followed by each extension stream in order of significance [Page 3, Fig. 3].)

Therefore, since *Kim* discloses the arrangement of a scalable stream of data by adding the component layers in order of significance, it would have been obvious to a person of ordinary skill in the art at the time of the invention to arrange the TS bit stream of The Standard in the order of layered significance by multiplexing the streams in the order of the significance of each layer in a particular timeframe. The motive to combine is to allow easy scalability by allowing the truncation of the bit stream at a particular enhancement layer simply by cutting off the stream after the desired layers have been received.

5. **Claims 4-7** are rejected under 35 U.S.C. 103(a) as being unpatentable over The ISO/IEC 13818-1 Standard ("The Standard") (Author Unknown, Generic Coding of Moving Pictures and Associated Audio: Systems, International Organization for Standardization, Workgroup 11 - Coding of Moving Pictures and Associate Audio, Pages 1-130, 13 November 1994) in view of *Kato*, et al. (US Patent No. 7,646,967 B2) and *Bruls*, et al., (US Pre Grant Publication No. 2006/0098937 A1).

Regarding claims 4, 6 and 7 *The Standard* discloses an information processing apparatus, method and computer to execute processing comprising:

a. Input means and step for inputting a stream including TS packets forming a first to nth elementary stream, and a TS packet storing a table in which information is written for associating IDs that respectively identify the TS packets with first to nth elementary stream formed of the TS packets (Pages xi-xix, 3-6, 10-13, 21-22,43-50). (The system of The Standard discloses a system for the encoding and transport of MPEG data [Pages xi-xix]. The system operates by receiving an incoming video stream [i.e. input stream] [See Fig. 0-1, "Video data" and "Audio Data", Page xi], encoding the stream into one or more packetized elementary streams ["PES"] associated with the input stream [Pages xi-xii], packetizing the one or more PES into a transport stream ["TS"] [Fig. 0-1, "Packetizer" and "Mux", Page xii] [See also Pages xvi-xix, particularly sections 0.4, 0.7 and 0.8], transmitting the TS across a network to a decoder [Page xii], receiving the TS at the decoder, separating out the input stream by de-multiplexing the PES's associated with the input stream, decoding the input stream and providing the decoded video/audio

for output [Pages xii-xiii, Particularly Fig. 0-2] [See also Pages xvi-xix, Particularly Sections 0.4, 0.7 and 0.8].)

b. A table stored in the TS packet input by the input means indicating the type of processable stream [Pages 35-36, “stream_ID” and Page 49, Table 2-29, “stream_type”). (The standard discloses that the stream type/stream ID of each elementary stream making up the processable stream/program is indicated using the stream_type or stream_ID [Pages 35-36, “stream_ID”] in the Program Map Table [Page 49]. For example, elementary streams containing 13813-3 audio streams have a stream type of “110x xxx” [Table 2-19, Page 35].)

c. Selecting means and step for selecting, from the stream, the TS packets having the ID associated with a selected stream (Page xiii). (The system of The Standard discloses a channel specific decoder, which de-multiplexes a particular channel/program [i.e. a "stream”] by determining the associated elementary streams using the program map table and extracting them from the TS [Pages xiii and Pages 48-49].)

d. Decoding means and step for decoding the TS packets selected by the selecting means (Pages xiii and Pages 48-49 - See (c), Supra).

The Standard fails to disclose determining if the system is capable of decoding a particular stream (i.e. if the stream is “processable”) before selecting and decoding a stream such that the information processing apparatus, method and program further comprises a determining step of referring to the table stored in the TS packet input by processing in the input

step and determining the type of processable stream and a selecting step of selecting, from the stream, the TS packets having the ID associated with the stream determined by processing in the determining step to be processable. In the same field of endeavor, *Kato* discloses determining if the system is capable of decoding a particular stream (i.e. if the stream is "processable") before selecting and decoding a stream such that the information processing apparatus, method and program further comprises a determining step of referring to the table stored in the TS packet input by processing in the input step and determining the type of processable stream and a selecting step of selecting, from the stream, the TS packets having the ID associated with the stream determined by processing in the determining step to be processable (Column 27, Lines 34-44). (The system of *Kato* discloses that the stream_ID can be used to determine the encoding type of the stream so that the system may determine if it is capable of decoding a stream before it initiates decoding of a stream).

Therefore, since *Kato* suggests the use of the stream_ID to determine if a stream may be decoded by the equipment, and The Standard discloses that the stream_ID is received in the program map table, it would have been obvious to a person of ordinary skill in the art at the time of the invention to check the decodeability of a particular stream by looking at the stream_ID of the member elementary streams before deciding to de-multiplex and decode the stream. The motive to combine is to prevent the system from attempting to decode a stream which it cannot decode.

The Standard fails to disclose that the first to nth elementary streams may comprise a base stream and a first to n-th extension streams such that the information processing apparatus, method and program comprises an input means or step for inputting a stream including TS packets forming a base stream, TS packets forming each of first to n-th extension streams having extensibility for the base stream, and a TS packet storing a table in which

information is written for associating IDs that respectively identify the TS packets with the base stream or the first to n-th extension streams formed of the TS packets. In the same field of endeavor, *Bruls* discloses that the first to nth elementary streams may comprise a base stream and a first to n-th extension streams such that the information processing apparatus, method and program comprises an input means or step for inputting a stream including TS packets forming a base stream, TS packets forming each of first to n-th extension streams having extensibility for the base stream, and a TS packet storing a table in which information is written for associating IDs that respectively identify the TS packets with the base stream or the first to n-th extension streams formed of the TS packets (Paragraphs 0031, 0039, and 0042). (The system of *Bruls* discloses the use of a base and one or more extensible enhancement layers for encoding and transporting programs using MPEG encoding techniques [Paragraphs 0031 and 0039, See also Paragraph 0003]. *Bruls* further discloses that the system uses a separate PID to identify the base and enhancement layers [Paragraph 0042].)

Therefore, since *Bruls* discloses the use of a base layer and multiple enhancement layers and the use of a separate PID for each of the layers, and *The Standard* discloses the use of a separate elementary stream for each PID for different encoding layers and the use of a program map table to identify the association of each PID/elementary stream with a particular program, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the PID identified layers of *Bruls* with the system of *The Standard* by associating each of the base and enhancement layers with a PID, as taught by *Bruls*, assigning each of the PIDs to a particular elementary stream, as taught by *The Standard*, and transmitting a table identifying the PIDs in a transport stream packet, as taught by *The Standard*. The motive to combine is to allow the system of *The Standard* to support layered video, which allows

increased flexibility and reduced bandwidth requirements for video distribution (See Generally, *Bruls*, Paragraphs 0006-0014).

Finally, assuming arguendo, that *Bruls* fails to disclose the use of more the one enhancement layers (i.e. Because *Bruls* is not in standard US format with a clearly identified background of the invention, it is unclear if Paragraph 0003 is a part of the background of the invention, therefore constituting a separate disclosure.) In the same field of endeavor, The Background of *Bruls* discloses the use of one or more enhancement layers (See Paragraph 0003).

Therefore, since the background of *Bruls* discloses the use of more then one extension layer (i.e. a first to n-the extension stream), it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the multiple extension layers of The Background of *Bruls* into the teachings of *The Specification* as modified by *Bruls* by transmitting more then one extension stream as a separate elementary stream with its own stream ID. The motive to combine is to allow enhanced flexibility by providing multiple extension streams so that endpoints can decode one or more of the extension streams to obtain incremental quality increases.

Regarding claim 5, *The Standard* discloses an information processing apparatus further comprising a buffering means for buffering, with respect to each ID, the TS packets selected by the selecting means (Fig. 2-6, “MB1..n” and “Bn” and “Bn” and "MBn”, Page 13). (The Standard discloses that each video and audio elementary stream has it s own separate buffer [Fig. 2-6, “MB1..n” and “Bn” and “Bn” and "MBn”, Page 13].)

6. **Claims 10, 14, 15, 16, 19 and 20** are rejected under 35 U.S.C. 103(a) as being unpatentable over The MPEG-4 Book (F. Pereira and T. Ebrahimi, The MPEG-4 Book, Prentice Hall, 10 July 2002, Pages 1-50) in view of *Herpel*, et al. (C. Herpel, A. Eleftheriadis, MPEG-4 Systems: Elementary Stream Management, Elsevier Science, Signal Processing: Image Communication, Vol. 15, 2000, Pages 299-320).

Regarding claims 10, 14 and 15, The MPEG-4 Book discloses an information processing apparatus, method and computer to execute processing comprising:

a. Encoding means and step for encoding, of an entire stream that may include a base stream and first to n-th extension streams having extensibility for the base stream, at least the base stream (Sections 7.1, 7.2, 7.4 and 10.3). (The system of The MPEG-4 Book discloses the use and transport of MPEG-4 Video and Audio encoded and decoded by the MPEG-4 protocol [See Section 7.1, Particularly Fig. 7.1, - Disclosing the Various Sections/Means for carrying out the system functionality]. The MPEG-4 Book further discloses the use of a low bit rate set of streams comprising a base stream and one or more enhancement layer streams [See Section 10.3.6]. For example, The MPEG-4 Book discloses the use of a scalable CLEP speech encoder that uses a base and one or more enhancement layer packetized streams [See Section 10.3.6]. The system of The MPEG-4 book further discloses the use of flexible multiplexing to allow groups of low bitrate MPEG-4 streams to be transported over a MPEG-2 packetized elementary stream [PES] [See Section 7.4, Particularly 7.4.3.2 - "Especially for the case of PSs, which do not allow carrying more than one SL-packetized stream, as well as for the case of a set of low-bit-rate streams, it is necessary to introduce a second multiplex layer into

the PES so that a number of MPEG-4 ESs can share a single PES. The FlexMux introduced earlier in this chapter is used for that purpose."] The identity of each of the Elementary Streams within a FlexMux stream is maintained by assigning a FMC_Descriptor which records the correspondence of each of the elementary stream IDs of the FlexMux stream to the stream_id [if a program stream] or the PID [if a transport stream]. Therefore, the MPEG 4 book discloses that groups of low bitrate channels, which may include low bitrate base and enhancement layers of a speech encoder, can be grouped under a common FlexMux channel, which is represented by a common PID [i.e. "first ID"].)

b. First adding means and step for adding a same first ID to, among the base stream and the first to n-th extension streams, the stream encoded by the encoding means, the first ID being used to identify the entire stream (See Section 7.4.1.3). (The MPEG-4 Book discloses that the PID [i.e. the First ID] that is assigned to the grouped base and enhancement layers [See (a), Supra] is added to the head of each of the transport stream packets in the form of the Packet ID [PID] inserted into the header of each [See Section 7.4.1.3]).

c. Second adding means and step for adding a second ID to, among the base stream and the first to n-th extension streams, the stream encoded by the encoding means, the second ID being used to identify each of the base stream and the first to n- th extension streams (See Sections 7.4.1.3 and 7.4.3.4). (The MPEG-4 Book further discloses that each ES_ID is associated with a particular FlexMux Channel number [i.e. "the second ID"] using the FMC_Descriptor [7.4.3.4]. Each FlexMux packet includes a FlexMux

channel number used to identify the FlexMux channel and by extension the ES_ID of the stream [Section 7.2, Particularly Fig. 7.7])

d. Packetizing means for packetizing the base stream and the first to n-th extension streams, to which the first ID and the second ID are added by the first adding means and the second adding means, into TS packets (See (a)-(c), Supra).

Assuming, arguendo, that The MPEG-4 Book fails to disclose the addition of the base and enhancement layers of an audio stream in a single FlexMux Stream such that the system comprises an information processing apparatus, method and computer to execute processing comprising a first adding means and step for adding a same first ID to, among the base stream and the first to n-th extension streams, the stream encoded by the encoding means and step, the first ID being used to identify the entire stream and a second adding means and step for adding a second ID to, among the base stream and the first to n-th extension streams, the stream encoded by the encoding means and step, the second ID being used to identify each of the base stream and the first to n-th extension streams. In the same field of endeavor, *Herpel* discloses the addition of the base and enhancement layers of a speech stream in a single FlexMux Stream such that the system comprises an information processing apparatus, method and computer to execute processing comprising a first adding means and step for adding a same first ID to, among the base stream and the first to n-th extension streams, the stream encoded by the encoding means and step, the first ID being used to identify the entire stream and a second adding means and step for adding a second ID to, among the base stream and the first to n-th extension streams, the stream encoded by the encoding means and step, the second ID being used to identify each of the base stream and the first to n-th extension streams

(i.e. The system of The MPEG-4 book discloses that "a set of low-bit-rate streams" can be transported using a common FlexMux Stream [See The MPEG-4 Book, Section 7.4.3.2]. The Office maintains that a person of ordinary skill in the art would have recognized that the term "a set of low-bit-rate streams" includes the set of low bitrate base and enhancement layer streams of the low bitrate CLEP speech encoder. However, even if this is incorrect, *Herpel* discloses the grouping of associated MPEG speech [Such as CPEP Streams] streams under a single FlexMux Stream [See Section 5.1, First Paragraph - The FlexMux multiplexer may be used for "low bitrate, low delay" streams such as speech streams].)

Therefore, since *Herpel* suggests the transport of low bitrate speech streams using a single FlexMux stream and The MPEG-4 Book discloses a set of low bandwidth speech streams comprising encoded base and enhancement layers, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the FlexMux speech streams of *Herpel* with the speech base and enhancement layers of *Herpel* by combining the Base and Enhancement layers of a scalable CLEP audio stream in a single FlexMux Stream. The motive to combine is provided by *Herpel* and is to lower transport overhead and to allow for low latency audio streams [Section 5.1].

Regarding claims 16, 19 and 20, The MPEG-4 Book discloses an information processing apparatus, method and computer to execute processing comprising:

a. Input means and step for inputting an entire stream that may include TS packets forming a base stream and TS packets forming each of first to n-th extension streams having extensibility for the base stream (Sections 7.1, 7.2, 7.4 and 10.3). (The system of The MPEG-4 Book discloses the use and transport of MPEG-4 Video and Audio encoded and decoded by the MPEG-4 protocol [See Section 7.1, Particularly Fig. 7.1, -

Disclosing the Various Sections/Means for carrying out the system functionality]. The MPEG-4 Book further discloses the use of a low bit rate set of streams comprising a base stream and one or more enhancement layer streams [See Section 10.3.6]. For example, The MPEG-4 Book discloses the use of a scalable CLEP speech encoder that uses a base and one or more enhancement layer packetized streams [See Section 10.3.6]. The system of The MPEG-4 book further discloses the use of flexible multiplexing to allow groups of low bitrate MPEG-4 streams to be transported over a MPEG-2 packetized elementary stream [PES] [See Section 7.4, Particularly 7.4.3.2 - "Especially for the case of PSs, which do not allow carrying more than one SL-packetized stream, as well as for the case of a set of low-bit-rate streams, it is necessary to introduce a second multiplex layer into the PES so that a number of MPEG-4 ESs can share a single PES. The FlexMux introduced earlier in this chapter is used for that purpose."] The identity of each of the Elementary Streams within a FlexMux stream is maintained by assigning a FMC_Descriptor which records the correspondence of each of the elementary stream IDs of the FlexMux stream to the stream_id [if a program stream] or the PID [if a transport stream]. Therefore, the MPEG 4 book discloses that groups of low bitrate channels, which may include low bitrate base and enhancement layers of an audio encoder, can be grouped under a common FlexMux channel, which is represented by a common PID [i.e. "first ID"], transmitted and received and decoded at a decoder. The MPEG-4 Book further discloses that each ES_ID is associated with a particular FlexMux Channel number [i.e. "the second ID"] using the FMC_Descriptor [7.4.3.4]. Each FlexMux packet includes a FlexMux channel number used to identify the FlexMux channel and by extension the ES_ID of the stream [Section 7.2, Particularly Fig. 7.7])

b. Selecting means and step for selecting, from the entire stream, the processable TS packets based on a first ID used to identify the entire stream, a second ID identifying each of the base stream and the first to n-th extension streams, and a predetermined condition set in advance, the first ID and the second ID being stored in each of the TS packets input by the input means (See Section 7.4.1.3). (The MPEG-4 Book discloses that the PID [i.e. the First ID] that is assigned to the grouped base and enhancement layers [See (a), Supra] is added to the head of each of the transport stream packets in the form of the Packet ID [PID] inserted into the header of each [See Section 7.4.1.3]. The MPEG-4 Book further discloses that the terminal device may select a particular program, which is identified by the associated PID and [if FlexMux is in use] the associated FlexMux Channels matched to each of the Elementary Streams for the program received via the program map [See Section 7.4.1.3 - Showing the Standard MPEG-2 Program Stream Program Map] [See also Section 7.4.3.4 - Showing the MPEG-4 FMC_Descriptor Extension, Which allows the association of the channels a FlexMux Stream assigned a single PID with individual Elementary Stream IDs [ES_ID]] by removing and decoding the PID(s) associated with the program from the packet stream [See Section 7.4.3.5, particularly Fig. 7.15].)

c. Decoding means and step for decoding the TS packets selected by the selecting means (Section 7.4, See (a) and (b), Supra).

Assuming, arguendo, that The MPEG-4 Book fails to disclose the addition of the base and enhancement layers of an audio stream in a single FlexMux Stream such that the system

comprises a selecting means and step for selecting, from the entire stream, the processable TS packets based on a first ID used to identify the entire stream, a second ID identifying each of the base stream and the first to n-th extension streams, and a predetermined condition set in advance, the first ID and the second ID being stored in each of the TS packets input by the input means. In the same field of endeavor, *Herpel* discloses the addition of the base and enhancement layers of an audio stream in a single FlexMux Stream such that the system comprises a selecting means and step for selecting, from the entire stream, the processable TS packets based on a first ID used to identify the entire stream, a second ID identifying each of the base stream and the first to n-th extension streams, and a predetermined condition set in advance, the first ID and the second ID being stored in each of the TS packets input by the input means (i.e. The system of The MPEG-4 book discloses that "a set of low-bit-rate streams" can be transported using a common FlexMux Stream [See The MPEG-4 Book, Section 7.4.3.2]. The Office maintains that a person of ordinary skill in the art would have recognized that the term "a set of low-bit-rate streams" includes the set of low bitrate base and enhancement layer streams of the low bitrate CLEP speech encoder. However, even if this is incorrect, *Herpel* discloses the grouping of associated MPEG speech [Such as CPEP Streams] streams under a single FlexMux Stream [See Section 5.1, First Paragraph - The FlexMux multiplexer may be used for "low bitrate, low delay" streams such as speech streams].)

Therefore, since *Herpel* suggests the transport of low bitrate speech streams using a single FlexMux stream and The MPEG-4 Book discloses a set of low bandwidth speech streams comprising encoded base and enhancement layers, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the FlexMux speech streams of *Herpel* with the speech base and enhancement layers of *Herpel* by combining the Base and Enhancement layers of a scalable CLEP audio stream in a single FlexMux Stream. The motive

to combine is provided by *Herpel* and is to lower transport overhead and to allow for low latency audio streams [Section 5.1].

7. **Claim 11 is** rejected under 35 U.S.C. 103(a) as being unpatentable over The MPEG-4 Book (F. Pereira and T. Ebrahimi, The MPEG-4 Book, Prentice Hall, 10 July 2002, Pages 1-50) in view of *Herpel*, et al. (C. Herpel, A. Eleftheriadis, MPEG-4 Systems: Elementary Stream Management, Elsevier Science, Signal Processing: Image Communication, Vol. 15, 2000, Pages 299-320) as applied to claim 10 and further in view of *Kim*, et al. (S. Kim, S. Park, Y. Kim, Fine Grain Scalability in MPEG-4 Audio, Audio Engineering Society, 111th Convention of The AES, 24 Sept 2001, Pages 1-5).

Regarding claim 11, The MPEG-4 Book fails to disclose a data structure wherein the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams, which are included in the entire stream, are arranged in sequence of the TS packets to be played back at the same time and in the order of the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams. In the same field of endeavor, *Kim* discloses a data structure wherein the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams, which are included in the entire stream, are arranged in sequence of the TS packets to be played back at the same time and in the order of the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams (Page 3, Fig. 3). (The system of Kim discloses a system where the base and enhancement layers of a transmitted bit stream for a timeslot/frame are multiplexed in layer order with the base stream first, followed by each extension stream in order of significance [Page 3, Fig. 3].)

Therefore, since *Kim* discloses the arrangement of a scalable stream of data by adding the component layers in order of significance, it would have been obvious to a person of ordinary skill in the art at the time of the invention to arrange the TS bit stream of The MPEG-4 Book in the order of layered significance by multiplexing the streams in the order of the significance of each layer in a particular timeframe. The motive to combine is to allow easy scalability by allowing the truncation of the bit stream at a particular enhancement layer simply by cutting off the stream after the desired layers have been received.

8. **Claims 12, 13 and 18** are rejected under 35 U.S.C. 103(a) as being unpatentable over The MPEG-4 Book (F. Pereira and T. Ebrahimi, The MPEG-4 Book, Prentice Hall, 10 July 2002, Pages 1-50) in view of *Herpel*, et al. (C. Herpel, A. Eleftheriadis, MPEG-4 Systems: Elementary Stream Management, Elsevier Science, Signal Processing: Image Communication, Vol. 15, 2000, Pages 299-320) as applied to claims 10 and 16 and further in view of *Wu*, et al. (US Patent No. 6,614,936).

Regarding claim 12, The MPEG-4 Book discloses the encoding means encodes the base stream and the extension streams into TS packets [See Section 7.4 and claim 1, supra]. The MPEG-4 Book further discloses that the session layer/DIMF Application Interface of the protocol stack is responsible for passing the data to be transmitted to the FlexMux layer via SL Packetized streams corresponding to each elementary stream (See Section 2.3, Particularly Fig. 2.3). The MPEG-4 Book fails to disclose when any of synchronization units of the first to n-th extension streams corresponding to synchronization units of the base stream are present, the encoding means encodes, among the first to n-th extension streams, the extension stream having the present synchronization units and the base stream. In the same field of endeavor,

Wu discloses when any of synchronization units of the first to n-th extension streams corresponding to synchronization units of the base stream are present, the encoding means encodes, among the first to n-th extension streams, the extension stream having the present synchronization units and the base stream (Column 2, Lines 10-49). (The system of *Wu* discloses a coder that uses a variable number of fine grain enhancement layers based on the available network bandwidth to encode and transmit data to a remote receiver [Column 1, Line 50 to Column 2, Line 49].)

Therefore, since *Wu* discloses encoding a variable number of enhancement layers according to available bandwidth and The MPEG-4 Book discloses that only data to be encoded is passed on to the FlexMux/Encoder layer, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the variable enhancement layers of *Wu* into the teachings of The MPEG-4 Book to create a system that encodes a variable number of enhancement layers based on available bandwidth. (i.e. That is, *Wu* discloses the encoding of a variable number of enhancement layers for transmission. However, at least arguably, *Wu* fails to disclose that encoding layer receives [i.e. receives "synchronization units" of] only the enhancement layers to be transmitted. The MPEG-4 Book provides this missing element, by showing the protocol stack of the terminal and demonstrating that only SL Packetized streams containing data to be transmitted are passed down to the FlexMux/Encoding Layer [See Section 2.3, Particularly Fig. 2.3] which encodes and transmits the received layers [See Sections 7.2 and 7.4].) The motive to combine is provided by *Wu* and is to allow for a variable number of enhancement layers in accordance with the available bandwidth of the channel, thereby allowing maximum transmission quality for a given channel (See Paragraph 0013).

Regarding claim 13, The MPEG-4 Book discloses the encoding means encodes the base stream and the extension streams into TS packets [See Section 7.4 and claim 1, supra].

The MPEG-4 Book further discloses that the session layer/DIMF Application Interface of the protocol stack is responsible for passing the data to be transmitted to the FlexMux layer via SL Packetized streams corresponding to each elementary stream (See Section 2.3, Particularly Fig. 2.3). The MPEG-4 Book fails to disclose wherein when any of the synchronization units of the first to n-th extension streams corresponding to the synchronization units of the base stream are present, the encoding means encodes, among the first to n-th extension streams, the extension stream having the present synchronization units and the base stream and does not encode the extension stream having none of the present synchronization units, thereby encoding the entire stream using variable bit rate. In the same field of endeavor, *Wu* discloses wherein when any of the synchronization units of the first to n-th extension streams corresponding to the synchronization units of the base stream are present, the encoding means encodes, among the first to n-th extension streams, the extension stream having the present synchronization units and the base stream and does not encode the extension stream having none of the present synchronization units, thereby encoding the entire stream using variable bit rate (Column 2, Lines 10-49). (The system of *Wu* discloses a coder that uses a variable number of fine grain enhancement layers based on the available network bandwidth to encode and transmit data to a remote receiver [Column 1, Line 50 to Column 2, Line 49].)

Therefore, since *Wu* discloses encoding a variable number of enhancement layers according to available bandwidth and The MPEG-4 Book discloses that only data to be encoded is passed on to the FlexMux/Encoder layer, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the variable enhancement layers of *Wu* into the teachings of The MPEG-4 Book to create a system that encodes a variable number of enhancement layers based on available bandwidth. (i.e. That is, *Wu* discloses the encoding of a variable number of enhancement layers for transmission. However, at least arguably, *Wu* fails to

disclose that encoding layer receives [i.e. receives "synchronization units" of] only the enhancement layers to be transmitted. The MPEG-4 Book provides this missing element, by showing the protocol stack of the terminal and demonstrating that only SL Packetized streams containing data to be transmitted are passed down to the FlexMux/Encoding Layer [See Section 2.3, Particularly Fig. 2.3] which encodes and transmits the received layers [See Sections 7.2 and 7.4].) The motive to combine is provided by *Wu* and is to allow for a variable number of enhancement layers in accordance with the available bandwidth of the channel, thereby allowing maximum transmission quality for a given channel (See Paragraph 0013).

Regarding claim 18, The MPEG-4 Book discloses the encoding means encodes the base stream and the extension streams into TS packets [See Section 7.4 and claim 1, supra]. The MPEG-4 Book further discloses that the session layer/DIMF Application Interface of the protocol stack is responsible for passing the data to be transmitted to the FlexMux layer via SL Packetized streams corresponding to each elementary stream (See Section 2.3, Particularly Fig. 2.3). The MPEG-4 Book fails to disclose an information processing apparatus wherein the entire stream input to the input means at least includes the encoded base stream, and further includes the first to n-th extension streams which correspond to synchronization units of the base stream and which are encoded using variable bit rate. In the same field of endeavor, *Wu* discloses an information processing apparatus wherein the entire stream input to the input means at least includes the encoded base stream, and further includes the first to n-th extension streams which correspond to synchronization units of the base stream and which are encoded using variable bit rate (Column 2, Lines 10-49). (The system of *Wu* discloses a coder that uses a variable number of fine grain enhancement layers based on the available network bandwidth to encode and transmit data to a remote receiver [Column 1, Line 50 to Column 2, Line 49].)

Therefore, since *Wu* discloses encoding a variable number of enhancement layers according to available bandwidth and The MPEG-4 Book discloses that only data to be encoded is passed on to the FlexMux/Encoder layer, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the variable enhancement layers of *Wu* into the teachings of The MPEG-4 Book to create a system that encodes a variable number of enhancement layers based on available bandwidth. (i.e. That is, *Wu* discloses the encoding of a variable number of enhancement layers for transmission. However, at least arguably, *Wu* fails to disclose that encoding layer receives [i.e. receives "synchronization units" of] only the enhancement layers to be transmitted. The MPEG-4 Book provides this missing element, by showing the protocol stack of the terminal and demonstrating that only SL Packetized streams containing data to be transmitted are passed down to the FlexMux/Encoding Layer [See Section 2.3, Particularly Fig. 2.3] which encodes and transmits the received layers [See Sections 7.2 and 7.4].) The motive to combine is provided by *Wu* and is to allow for a variable number of enhancement layers in accordance with the available bandwidth of the channel, thereby allowing maximum transmission quality for a given channel (See Paragraph 0013).

9. **Claim 17** is rejected under 35 U.S.C. 103(a) as being unpatentable over The MPEG-4 Book (F. Pereira and T. Ebrahimi, The MPEG-4 Book, Prentice Hall, 10 July 2002, Pages 1-50) and *Herpel*, et al. (C. Herpel, A. Eleftheriadis, MPEG-4 Systems: Elementary Stream Management, Elsevier Science, Signal Processing: Image Communication, Vol. 15, 2000, Pages 299-320) as applied to claim 16 and further in view of *Kim*, et al. S. Kim, S. Park, Y. Kim, Fine Grain Scalability in MPEG-4 Audio, Audio Engineering Society, 111th Convention of The AES, 24 Sept 2001, Pages 1-5).

Regarding claim 17, The MPEG-4 Book fails to disclose an information processing apparatus wherein to the input means, the entire stream is input, including the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams arranged in sequence of the TS packets to be played back at the same time and in the order of the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams. In the same field of endeavor, *Kim* discloses an information processing apparatus wherein to the input means, the entire stream is input, including the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams arranged in sequence of the TS packets to be played back at the same time and in the order of the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams (Page 3, Fig. 3). (The system of Kim discloses a system where the base and enhancement layers of a transmitted bit stream for a timeslot/frame are multiplexed in layer order with the base stream first, followed by each extension stream in order of significance [Page 3, Fig. 3].)

Therefore, since *Kim* discloses the arrangement of a scalable stream of data by adding the component layers in order of significance, it would have been obvious to a person of ordinary skill in the art at the time of the invention to arrange the TS bit stream of The MPEG-4 Book in the order of layered significance by multiplexing the streams in the order of the significance of each layer in a particular timeframe. The motive to combine is to allow easy scalability by allowing the truncation of the bit stream at a particular enhancement layer simply by cutting off the stream after the desired layers have been received.

10. **Claim 21** is rejected under 35 U.S.C. 103(a) as being unpatentable over The MPEG-4 Book (F. Pereira and T. Ebrahimi, The MPEG-4 Book, Prentice Hall, 10 July 2002, Pages 1-50)

in view of *Herpel*, et al. (C. Herpel, A. Eleftheriadis, MPEG-4 Systems: Elementary Stream Management, Elsevier Science, Signal Processing: Image Communication, Vol. 15, 2000, Pages 299-320) and *Wu*, et al. (US Patent No. 6,614,936).

Regarding claim 21, The MPEG-4 Book discloses a data structure of an entire stream to be played back by a computer, wherein the entire stream may include a base stream and first to n-th extension streams having extensibility for the base stream, the entire stream includes:

a. TS packets forming the base stream, TS packets forming, among the first to n-th extension streams, the extension streams (Sections 7.1, 7.2, 7.4 and 10.3). (The system of The MPEG-4 Book discloses the use and transport of MPEG-4 Video and Audio encoded and decoded by the MPEG-4 protocol [See Section 7.1, Particularly Fig. 7.1, - Disclosing the Various Sections/Means for carrying out the system functionality]. The MPEG-4 Book further discloses the use of a low bit rate set of streams comprising a base stream and one or more enhancement layer streams [See Section 10.3.6]. For example, The MPEG-4 Book discloses the use of a scalable CLEP speech encoder that uses a base and one or more enhancement layer packetized streams [See Section 10.3.6]. The system of The MPEG-4 book further discloses the use of flexible multiplexing to allow groups of low bitrate MPEG-4 streams to be transported over a MPEG-2 packetized elementary stream [PES] [See Section 7.4, Particularly 7.4.3.2 - "Especially for the case of PSs, which do not allow carrying more than one SL-packetized stream, as well as for the case of a set of low-bit-rate streams, it is necessary to introduce a second multiplex layer into the PES so that a number of MPEG-4 ESs can share a single PES. The FlexMux introduced earlier in this chapter is used for that

purpose."] The identity of each of the Elementary Streams within a FlexMux stream is maintained by assigning a FMC_Descriptor which records the correspondence of each of the elementary stream IDs of the FlexMux stream to the stream_id [if a program stream] or the PID [if a transport stream]. Therefore, the MPEG 4 book discloses that groups of low bitrate channels, which may include low bitrate base and enhancement layers of an audio encoder, can be grouped under a common FlexMux channel, which is represented by a common PID [i.e. "first ID"], transmitted and received and decoded at a decoder. The MPEG-4 Book further discloses that each ES_ID is associated with a particular FlexMux Channel number [i.e. "the second ID"] using the FMC_Descriptor [7.4.3.4]. Each FlexMux packet includes a FlexMux channel number used to identify the FlexMux channel and by extension the ES_ID of the stream [Section 7.2, Particularly Fig. 7.7])

b. A header of each of the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams includes a first ID used to identify the entire stream and a second ID identifying each of the base stream and the first to n-th extension streams (See (a), Supra).

Assuming, arguendo, that The MPEG-4 Book fails to disclose the addition of the base and enhancement layers of an audio stream in a single FlexMux Stream such that the system comprises a selecting means and step for selecting, from the entire stream, the processable TS packets based on a first ID used to identify the entire stream, a second ID identifying each of the base stream and the first to n-th extension streams, and a predetermined condition set in advance, the first ID and the second ID being stored in each of the TS packets input by the input

means. In the same field of endeavor, *Herpel* discloses the addition of the base and enhancement layers of an audio stream in a single FlexMux Stream such that the system comprises a selecting means and step for selecting, from the entire stream, the processable TS packets based on a first ID used to identify the entire stream, a second ID identifying each of the base stream and the first to n-th extension streams, and a predetermined condition set in advance, the first ID and the second ID being stored in each of the TS packets input by the input means (i.e. The system of The MPEG-4 book discloses that "a set of low-bit-rate streams" can be transported using a common FlexMux Stream [See The MPEG-4 Book, Section 7.4.3.2]. The Office maintains that a person of ordinary skill in the art would have recognized that the term "a set of low-bit-rate streams" includes the set of low bitrate base and enhancement layer streams of the low bitrate CLEP speech encoder. However, even if this is incorrect, *Herpel* discloses the grouping of associated MPEG speech [Such as CPEP Streams] streams under a single FlexMux Stream [See Section 5.1, First Paragraph - The FlexMux multiplexer may be used for "low bitrate, low delay" streams such as speech streams].)

Therefore, since *Herpel* suggests the transport of low bitrate speech streams using a single FlexMux stream and The MPEG-4 Book discloses a set of low bandwidth speech streams comprising encoded base and enhancement layers, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the FlexMux speech streams of *Herpel* with the speech base and enhancement layers of *Herpel* by combining the Base and Enhancement layers of a scalable CLEP audio stream in a single FlexMux Stream. The motive to combine is provided by *Herpel* and is to lower transport overhead and to allow for low latency audio streams [Section 5.1].

The MPEG-4 Book further discloses that the session layer/DIMF Application Interface of the protocol stack is responsible for passing the data to be transmitted to the FlexMux layer via

SL Packetized streams corresponding to each elementary stream (See Section 2.3, Particularly Fig. 2.3). The MPEG-4 Book fails to disclose TS packets forming, when any of synchronization units of the first to n-th extension streams corresponding to synchronization units of the base stream are present, among the first to n-th extension streams, the extension stream having the present synchronization units. In the same field of endeavor, *Wu* discloses TS packets forming, when any of synchronization units of the first to n-th extension streams corresponding to synchronization units of the base stream are present, among the first to n-th extension streams, the extension stream having the present synchronization units (Column 2, Lines 10-49). (The system of *Wu* discloses a coder that uses a variable number of fine grain enhancement layers based on the available network bandwidth to encode and transmit data to a remote receiver [Column 1, Line 50 to Column 2, Line 49].)

Therefore, since *Wu* discloses encoding a variable number of enhancement layers according to available bandwidth and The MPEG-4 Book discloses that only data to be encoded is passed on to the FlexMux/Encoder layer, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the variable enhancement layers of *Wu* into the teachings of The MPEG-4 Book to create a system that encodes a variable number of enhancement layers based on available bandwidth. (i.e. That is, *Wu* discloses the encoding of a variable number of enhancement layers for transmission. However, at least arguably, *Wu* fails to disclose that encoding layer receives [i.e. receives “synchronization units” of] only the enhancement layers to be transmitted. The MPEG-4 Book provides this missing element, by showing the protocol stack of the terminal and demonstrating that only SL Packetized streams containing data to be transmitted are passed down to the FlexMux/Encoding Layer [See Section 2.3, Particularly Fig. 2.3] which encodes and transmits the received layers [See Sections 7.2 and 7.4].) The motive to combine is provided by *Wu* and is to allow for a variable number of

enhancement layers in accordance with the available bandwidth of the channel, thereby allowing maximum transmission quality for a given channel (See Paragraph 0013).

Regarding claim 23, The MPEG-4 Book fails to disclose a data structure according wherein the entire stream at least includes the base stream, and further includes the TS packets forming the first to n- th extension streams corresponding to the synchronization units of the base stream, the number of the TS packets being variable. In the same field of endeavor, *Wu* discloses a data structure according wherein the entire stream at least includes the base stream, and further includes the TS packets forming the first to n- th extension streams corresponding to the synchronization units of the base stream, the number of the TS packets being variable (Column 2, Lines 10-49). (The system of *Wu* discloses a coder that uses a variable number of fine grain enhancement layers based on the available network bandwidth to encode and transmit data to a remote receiver [Column 1, Line 50 to Column 2, Line 49].)

Therefore, since *Wu* discloses encoding a variable number of enhancement layers according to available bandwidth and The MPEG-4 Book discloses that only data to be encoded is passed on to the FlexMux/Encoder layer, it would have been obvious to a person of ordinary skill in the art at the time of the invention to implement the variable enhancement layers of *Wu* into the teachings of The MPEG-4 Book to create a system that encodes a variable number of enhancement layers based on available bandwidth. (i.e. That is, *Wu* discloses the encoding of a variable number of enhancement layers for transmission. However, at least arguably, *Wu* fails to disclose that encoding layer receives [i.e. receives "synchronization units" of] only the enhancement layers to be transmitted. The MPEG-4 Book provides this missing element, by showing the protocol stack of the terminal and demonstrating that only SL Packetized streams containing data to be transmitted are passed down to the FlexMux/Encoding Layer [See Section 2.3, Particularly Fig. 2.3] which encodes and transmits the received layers [See Sections 7.2

and 7.4].) The motive to combine is provided by *Wu* and is to allow for a variable number of enhancement layers in accordance with the available bandwidth of the channel, thereby allowing maximum transmission quality for a given channel (See Paragraph 0013).

11. **Claim 22** is rejected under 35 U.S.C. 103(a) as being unpatentable over The MPEG-4 Book (F. Pereira and T. Ebrahimi, The MPEG-4 Book, Prentice Hall, 10 July 2002, Pages 1-50), *Herpel*, et al. (C. Herpel, A. Eleftheriadis, MPEG-4 Systems: Elementary Stream Management, Elsevier Science, Signal Processing: Image Communication, Vol. 15, 2000, Pages 299-320) and *Wu*, et al. (US Patent No. 6,614,936) as applied to claim 21 and further in view of *Kim*, et al. S. Kim, S. Park, Y. Kim, Fine Grain Scalability in MPEG-4 Audio, Audio Engineering Society, 111th Convention of The AES, 24 Sept 2001, Pages 1-5).

Regarding claim 22, The MPEG-4 Book fails to disclose a data structure wherein the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams, which are included in the entire stream, are arranged in sequence of the TS packets to be played back at the same time and in the order of the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams. In the same field of endeavor, *Kim* discloses data structure wherein the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams, which are included in the entire stream, are arranged in sequence of the TS packets to be played back at the same time and in the order of the TS packets forming the base stream and the TS packets forming each of the first to n-th extension streams (Page 3, Fig. 3). (The system of Kim discloses a system where the base and enhancement layers of a transmitted bit stream for a timeslot/frame are

multiplexed in layer order with the base stream first, followed by each extension stream in order of significance [Page 3, Fig. 3].)

Therefore, since *Kim* discloses the arrangement of a scalable stream of data by adding the component layers in order of significance, it would have been obvious to a person of ordinary skill in the art at the time of the invention to arrange the TS bit stream of The MPEG-4 Book in the order of layered significance by multiplexing the streams in the order of the significance of each layer in a particular timeframe. The motive to combine is to allow easy scalability by allowing the truncation of the bit stream at a particular enhancement layer simply by cutting off the stream after the desired layers have been received.

Prior art Made of Record

12. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure:

a. IEEE 11496-3, 2001, Subpart 4: General Audio Coding (Author Unknown, IEEE 11496-3, Amendment 2, 2001, Subpart 4: General Audio Coding, ISO Press, 2001, Pages 1-87 and 307-311) - Disclosing detains of MPEG Audio Coding, including the BSAC Fine Grain Audio Scaling, which allows for the scaling of compressed audio in small increments, carried in interleaved base and enhancement layer elementary streams.

b. *Lee*, et al. (S. Lee, S. Kim, E. Oh, A real Time Audio Streaming Meathod for Time Varying Network Loads, 112th Convention of The AES, May 13, 2002, Pages 1-4) -

Disclosing a scalable fine grained audio transmission protocol for the transmission of MPEG-4 Audio over an IP Network.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Christopher Crutchfield whose telephone number is (571) 270-3989. The examiner can normally be reached on Monday through Friday 8:00 AM to 5:00 PM EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Daniel Ryman can be reached on (571) 272-3152. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Christopher Crutchfield/
Examiner, Art Unit 2466
6/3/2010

/Daniel J. Ryman/
Supervisory Patent Examiner, Art Unit 2466